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In re application of inventor(s):
Christoph Menzel et al.

Application No. **09/830,480**

Confirmation No. **8158**

Filing Date: **26 April 2001**

Title: **Internet Based Hearing Assessment Methods**

Group Art Unit: **2153**
Examiner: **Aaron N. Strange**

CUSTOMER NO. 22470

MAIL STOP AMENDMENT
Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

COMBINED DECLARATION OF CHRISTOPH MENZEL, SUNIL PURIA, R. SCOTT RADER AND VINCENT PLUVINAGE

We, Christoph Menzel, Sunil Puria, R. Scott Rader, and Vincent Pluvinage, declare as follows:

1. We are the co-inventors of the claims in the above-identified patent application (hereinafter the "instant application"). We conceived the invention that is the subject of the instant application, in the United States of America, prior to 24 January 2000, and diligently pursued reduction of the invention to practice, in the United States of America, as corroborated by the attached Exhibits, through at least 18 September 2000.

2. We are informed and believe that the instant application is a national phase filing of International Application No. PCT/US00/40931, and that such International Application was filed 18 September 2000.

3. Exhibit A, which is attached hereto, is an invention disclosure with the date being redacted showing several embodiments of the invention such as embodiments disclosed in Figures 3, 4, 5, 6 and 7 of the instant application, which was generated before, and establishes that, the invention was conceived before 24 January 2000 in the United States of America.

4. Exhibits B-G, which are attached hereto, corroborate diligent efforts, in the United States of America by the undersigned and under the direction of the undersigned, to reduce the invention to practice between 24 January 2000 and August 2000, inclusive, with Exhibit B being

a memorandum, dated February 4, 2000, concerning loudness matching test development that relates to the sound test embodiment disclosed in the patent application identified in paragraph 1; Exhibit C being a specification generated to facilitate performing an audiometric test using a standard personal computer and calibrated headphones in accordance with an embodiment of the invention disclosed in the patent application identified in paragraph 1 and which we are informed and believe is a printed copy of the specification saved in Microsoft® Visual Source Safe®, which is a software program that maintains an audit trail for the document, and that the document was generated no later than February 2000; Exhibit D being a copy of a memorandum, with the date being redacted, discussing the design of a web-based system to implement the invention disclosed the patent application identified in paragraph 1, which we are informed and believe is a printed copy of the memorandum saved employing Norton Ghost™, which enables generating back-ups of information stored on a computer, and that the document was generated no later than March 2000; Exhibit E being a memorandum that discusses a program by which to ascertain the functionality of the tests implemented by the invention disclosed in the patent application identified in paragraph 1, which we are informed and believe is a printed copy of the memorandum saved in Microsoft® Visual Source Safe®, which is a software program that maintains an audit trail for the document, and that the document was generated no later than April 2000; Exhibit F being a memorandum, dated May 26, 2000, discussing control of a sound processing device from a computer employing various peripheral devices in furtherance of the invention disclosed in the patent application identified in paragraph 1; and Exhibit G being a redacted copy of pages 66, 67 and 77-80 of Mr. Menzel's notes, dated June, July and August 2000 concerning various developmental tests in furtherance of developing the invention disclosed in the patent application identified in paragraph 1.

5. During the month of September 2000 we worked in cooperation with our patent attorney to facilitate the filing of the international patent application identified in paragraph 2 up to an including the international filing date of 18 September 2000.

The undersigned declares that all statements made herein of his own knowledge are true and that all statements made on information and belief are believed to be true; and further that the statements are made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code and that such willful false statements may jeopardize the validity of the application and any patent that may issue therefrom.

DATE: 6/4/06 Christoph Menzel

Christoph Menzel

DATE:________ Sunil Puria

DATE:________ R. Scott Rader

DATE:________ Vincent Pluvinage

Exhibit C being a specification generated to facilitate performing an audiometric test using a standard personal computer and calibrated headphones in accordance with an embodiment of the invention disclosed in the patent application identified in paragraph 1 and which we are informed and believe is a printed copy of the specification saved in Microsoft® Visual Source Safe®, which is a software program that maintains an audit trail for the document, and that the document was generated no later than February 2000; Exhibit D being a copy of a memorandum, with the date being redacted, discussing the design of a web-based system to implement the invention disclosed the patent application identified in paragraph 1, which we are informed and believe is a printed copy of the memorandum saved employing Norton Ghost™, which enables generating back-ups of information stored on a computer, and that the document was generated no later than March 2000; Exhibit E being a memorandum that discusses a program by which to ascertain the functionality of the tests implemented by the invention disclosed in the patent application identified in paragraph 1, which we are informed and believe is a printed copy of the memorandum saved in Microsoft® Visual Source Safe®, which is a software program that maintains an audit trail for the document, and that the document was generated no later than April 2000; Exhibit F being a memorandum, dated May 26, 2000, discussing control of a sound processing device from a computer employing various peripheral devices in furtherance of the invention disclosed in the patent application identified in paragraph 1; and Exhibit G being a redacted copy of pages 66, 67 and 77-80 of Mr. Menzel's notes, dated June, July and August 2000 concerning various developmental tests in furtherance of developing the invention disclosed in the patent application identified in paragraph 1.

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DATE:________________ ________________
Christoph Menzel

DATE: 6/2/2006 ________________
Sunil Puria

DATE:________________ ________________
R. Scott Rader

DATE:________________ ________________
Vincent Pluvinage

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DATE:__________ __________ Christoph Menzel

DATE:__________ __________ Sunil Puria

DATE: 5/30/2006 __________ R. Scott Rader

DATE:__________ __________ Vincent Pluvinage

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DATE:________________ ________________
Christoph Menzel

DATE:________________ ________________
Sunil Puria

DATE:________________ ________________
R. Scott Rader

DATE:________________ ________________
Vincent Pluvinage

EXHIBIT A

Rx Sound
Personalized Listening

MEMO: Invention disclosure for Internet Based Hearing Loss Assessment Tests
From: Christoph Menzel, Sunil Puria and R. Scott Rader

Date [redacted]

It is of commercial interest to develop a hearing loss assessment test that can be administered via the Internet using consumer quality Internet electronics eg. Personal computer, web TV. The following document discloses a number test types and a number of implement for Internet based Hearing Loss Assessment Tests. While the methods described below contain many differences, there are a few over arching similarities. The similarities are:

- Test protocol delivered to consumer via the internet
- Consumer's sound card is used for sound generation
- Test protocol assesses hearing loss across audible frequency range or any subset of the audible range
- Test is self-administered ie the consumer follows the test protocol without interaction with an expert.

Test Types

There are a number of different test types that can be used within an Internet based hearing loss assessment. The different types are dependent on the type of data. These criteria are usable with many different test configurations. The different test types are:

- Hearing threshold level
- Masking threshold level
- Loudness matching

The hearing threshold level test type is centered upon identifying the sound level when the test subject can just begin to hear the test signal. This test type may be associated with determining that actual SPL of threshold across the frequency range or the test method may be simply to establish the relative level of threshold as a function of frequency.

The masking threshold level test type identifies the test signal sound level when the test signal can be hear out of a masking signal. The masking threshold test protocol can be completed at a number of masking signal amplitudes to give an indication of recruitment.

This method may have some advantages when there is some background noise at frequencies other than the test frequency.

In the loudness matching method, the generated sound consists of two different frequencies. One frequency is considered a baseline and is constant throughout a test. The other sound, the test sound, has a variable amplitude and frequency during the test. A measurement consists of determining the loudness of the test sound that matches the loudness of the baseline sound at each of the test frequencies. The resulting measurements are used to generate an equal loudness curve. The difference between the equal loudness curve obtained here and the equal loudness curve for normal hearing populations gives the hearing loss assessment. This test protocol can be completed at a number of different baseline amplitudes to give an indication of recruitment. The loudness matching method as used for Internet hearing testing was invented by Sunil Puria.

The methods differ in terms of what extra equipment must be added to the basic computer in order to complete the test. The differences have implications with respect to test methodology, accuracy and test flow complexity. A drawing of the common system components and architecture is shown in figure 1.

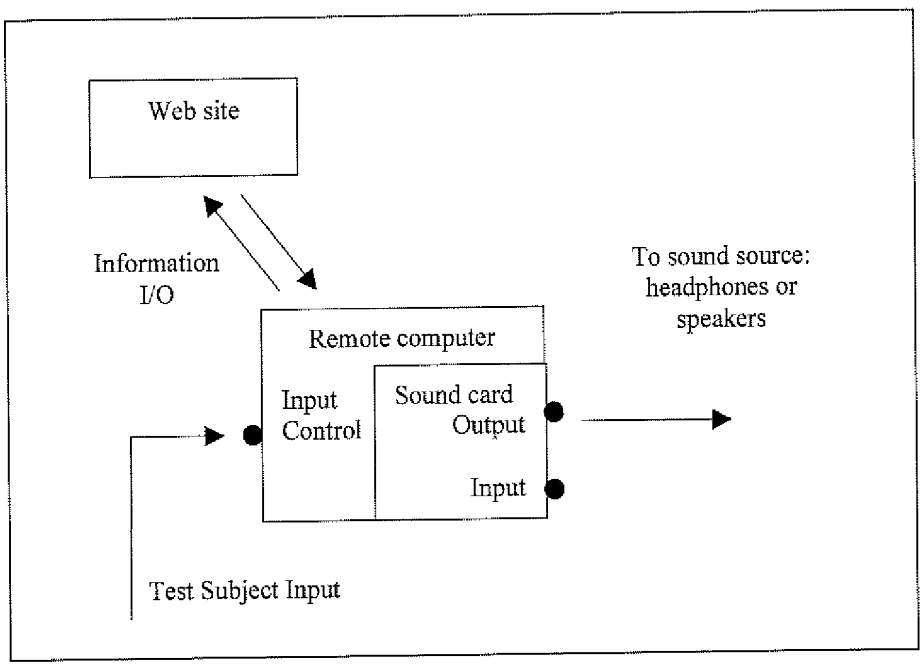


Figure #1

Test Type Implementation

Each of these test types can be implemented in a number of different test configurations and test conditions. The following text outlines the test configurations and conditions that can be used with the test types.

General Implementation Options

A number of implementation options are implicitly compatible with all of the methods disclosed below and hence should be considered as covered across all disclosed test methods.

Test Subject Input

There are numerous options for receiving the test subject's feedback. They are:

- Adjust volume control until some criteria is reached and then signal that the test step is complete through a keystroke, mouse click or a time out
- Complete an action when test generated sound meets some criteria eg Test sound is a tone varying in loudness. The test subject enters a mouse click when the sound disappears.

Test control/Data Processing

The test control can be either through the web link or from an executable, run locally on the test subject's equipment or any partitioning of control in between. Similarly, data collected during the test could be returned to the web site server as raw data, as the completely analyzed result, at any level in between or the data may not be returned to the web site server at all.

Test Sound Signals

The type of test signal used can have significant influence on the results of the test through a number of psychoacoustic effects. Hence a number of different test signal types will be of interest. The test methods outlined are not limited to specific types of test tones. Examples of the types of test tones that might be used are:

- Pure tones of long duration and constant intensity in each test step utilizing a number of different test steps at different frequencies
- Pulses of pure tones and constant intensity in each step utilizing a number of different test steps at different frequencies
- Combinations of tones of long duration and varying intensity in each test step utilizing a number of different test steps at different frequencies
- Pulses of combinations of tones of varying intensity in each step utilizing a number of different test steps at different frequencies
- Constant amplitude, swept frequency sound in each test step utilizing different test steps at different amplitudes
- Constant amplitude pulses of swept frequency sound in each test step utilizing different test steps at different amplitudes
- Bandpass filtered noise combined with test signals.

Furthermore the method of test sound signal generation is not limited and should include MIDI, synthesis or wavetables.

Monoaural or Binaural Implementation

The test methods outlined below could be implemented in either a monaural or a binaural configuration. In the monaural implementation, each ear is tested individually and the

other ear is "plugged" or otherwise deprived of test signal input. In an implementation scheme in which the headphones are supplied, the supplied headphones may have only one speaker.

Clearly, there are advantages and disadvantages associated with either implementation with respect to accuracy and test complexity.

Specific Test Configurations

The three basic test methods outlined above can be implemented within a number of different test set-ups. The test set-up will require different peripheral equipment, test protocols and they may have different levels of accuracy.

Four general test configurations are possible. The configurations are based on the four possible independent combinations of the following two independent test concepts

SPL actually measured (SPL_r) OR relative SPL levels used (SPL_r)
SPL_a or SPL_r measured at the ear OR SPL_a or SPL_r inferred from drive voltage measurements

Actual SPL Measurement Methods

Actual SPL measurement methods measure the actual sound pressure level that corresponds with the test criterion. As such, these measurements require measurement of a quantity that can be related to sound pressure level. Generally extra equipment is needed. Furthermore, the extra equipment will need to be calibrated equipment.

Actual SPL Measurement Inferred from Drive Voltage This general method refers to the approach of using a direct measurement of the sound sources drive voltage and knowledge of the sound source's voltage-to-sound transfer function to estimate the SPL delivered during the test. Since the actual sound pressure level delivered to the ear from a given source depends greatly on the relative geometry of the source and the ear, this test method probably requires that headphones generate the sound.

Calibration Source Method

In this configuration, a calibration source box is used in conjunction with calibrated headphones. The calibration source is used to determine the specific transfer functions of the sound card input A/D and output D/A conversions. Once the input A/D and output D/A calibrations are known, it is possible to accurately know the output drive signal. Combining this calibration knowledge with the known headphone calibration, the absolute sound pressure level can be estimated. The calibration source box is a three port box containing two switches and three switch contacts. It may be configured with its own power source through either a battery connection or through a wall plug.

The calibration and test process is as follows:

Step number	Switch 1	Switch 2	Sound Card Output	Sound Card Input	Measurement
1	A or B	C	No output	Cal. Source	Determines the transfer function of the input A/D.
2	B	B	Cal. Signal Suite	Test Signal	Using the results from step 1, determines the transfer function of the output A/D.
3 repeat @ all freq. values	A	B	Test Signal Suite	No input	Test subject signals when a test criteria is met Using data from step 1, the microphones calibration data and the microphone output, the corresponding SPL can be determined.

A simulated earphone load may need to be attached to connection point B during step two of the test to ensure that the output impedance seen by the sound card during calibration is the same as the load seen during the test. If this extra load is needed, then another switch and terminal will be needed.

The measured sound pressure levels are the sound pressure levels that meet the test criterion at the various test frequencies. These data are then used to calculate hearing loss according to the test type. The test set-up is shown in figure #2.

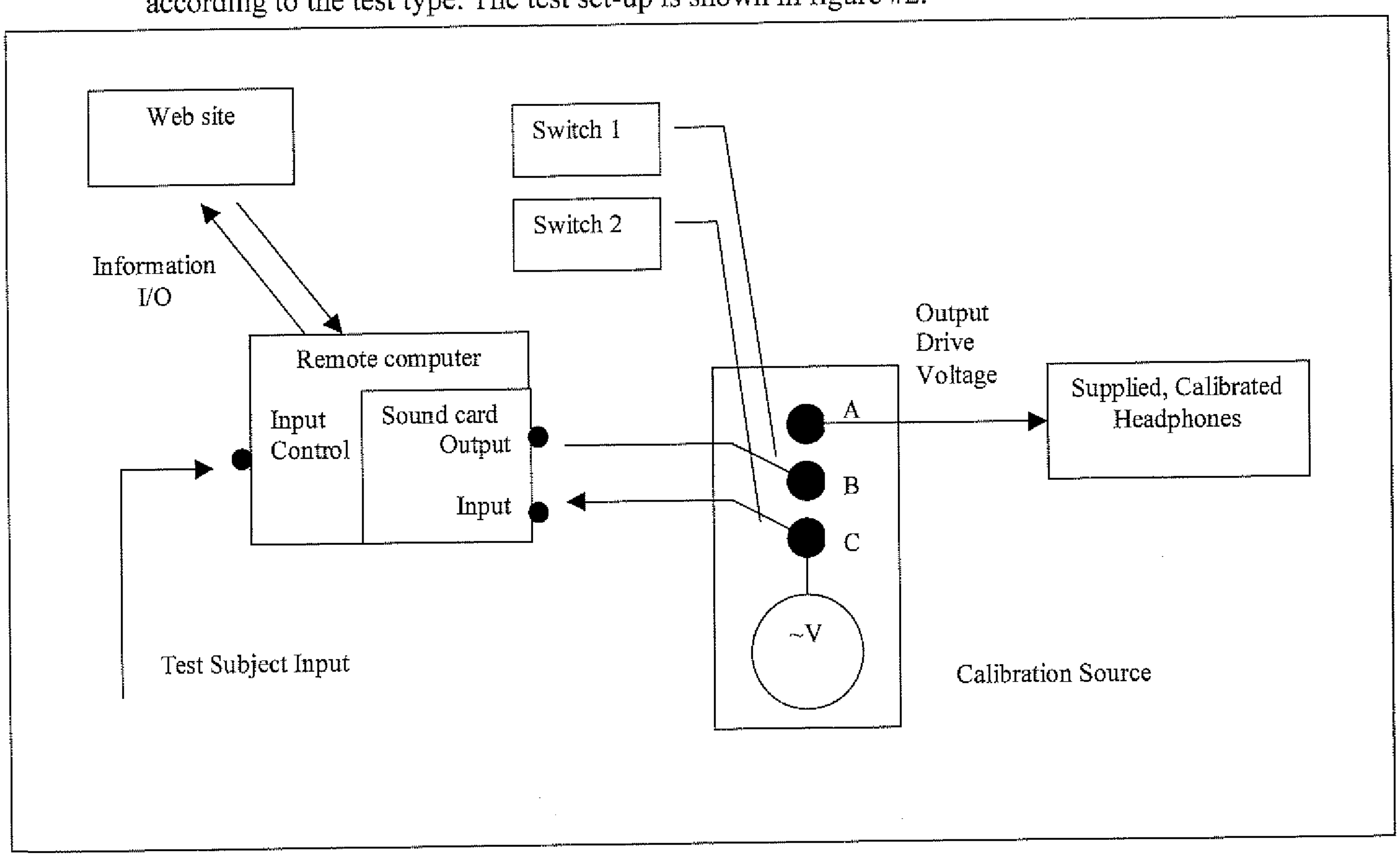


Figure #2

Note that if it is assumed that the sound card's input A/D transfer function is assumed to be one, then an actual SPL measurement can be made using the calibrated headphones but with out the calibration source box. In this case, calibration is not required and the headphone output can be directly fed into the input of the sound card.

The VCO Method This method assumes that the sound card input port's frequency–to-frequency transfer function is one. Using this assumption, a VCO is used to map the calibrated headphone drive voltage amplitude into frequency space. The drive voltage is determined through frequency space analysis of the VCO output combined with known headphone calibration. The test set-up is shown in Figure #3.

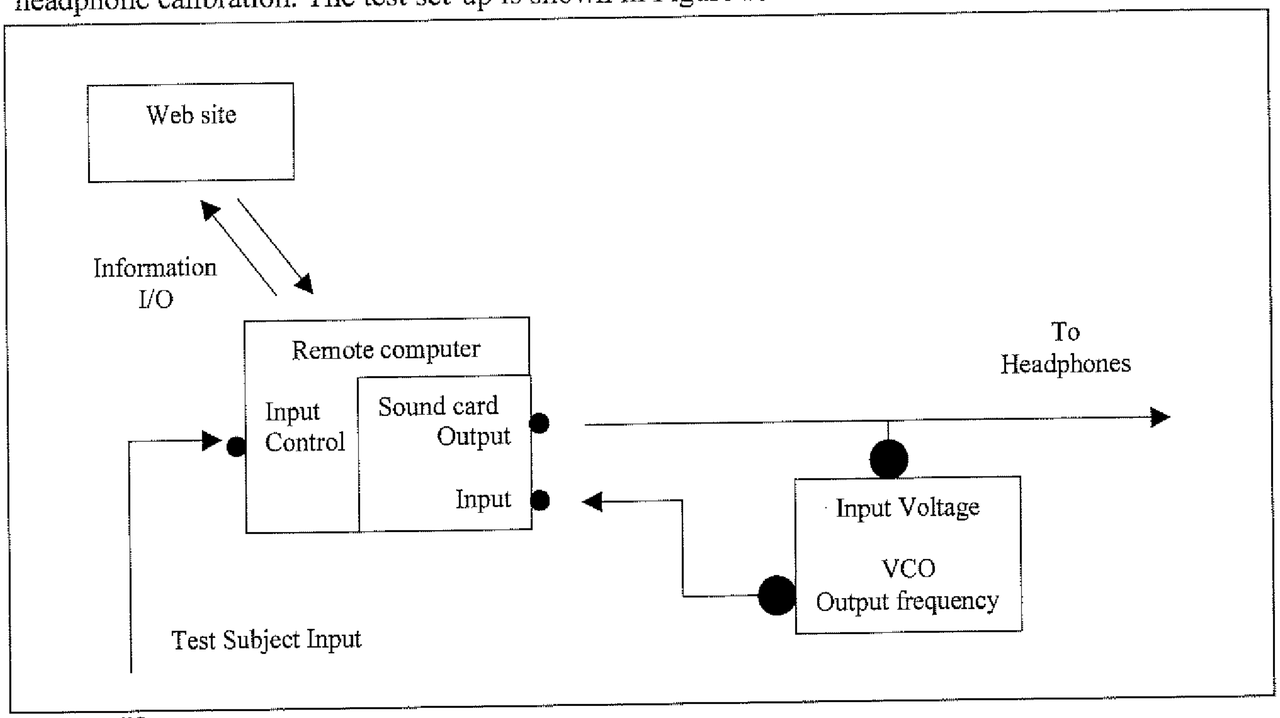


Figure #3

Using the measured frequency of the VCO output, the known VCO mapping function and the known headphone calibration, the corresponding SPL can be determined. The measured sound pressure levels at the various test frequencies gives the hearing loss assessment.

Actual SPL Measurement Completed at Ear

This general method uses direct measurement of the sound pressure level at the ear. Here a calibrated microphone is used to measure the sound pressure level during the test. Since a measurement of sound pressure is made at the ear, this test method is not limited to headphones.

Calibration Source Method

In this configuration, a calibration source box is used in conjunction with a calibrated microphone. The calibration source is used to determine the specific transfer functions of the sound card input A/ conversion. Combining this calibration knowledge with the known headphone calibration, the sound pressure level can be determined. The calibration source box is a four port box containing three switches and three switch contacts. It may be configured with its own power source through either a battery connection or through a wall plug. The test set-up is shown in figure #4.

The calibration and test process are shown in the following table.

Step number	Switch 1	Switch 2	Switch 3	Sound Card Output	Sound Card Input	Measurement
1	A	C	B	No output	Calibration Source	Determines the transfer function of the input A/D.
2 repeat @ all freq. values	A	B	C	Test Signal Suite	SPL microphon e	Test subject signals when a test criterion is met. Using data from step 1, the microphone calibration data and microphone output, the corresponding SPL can be determined.

The measured sound pressure levels are the sound pressure levels that meet the test criteria at the various test frequencies. These data are then used to calculate hearing loss according the test type.

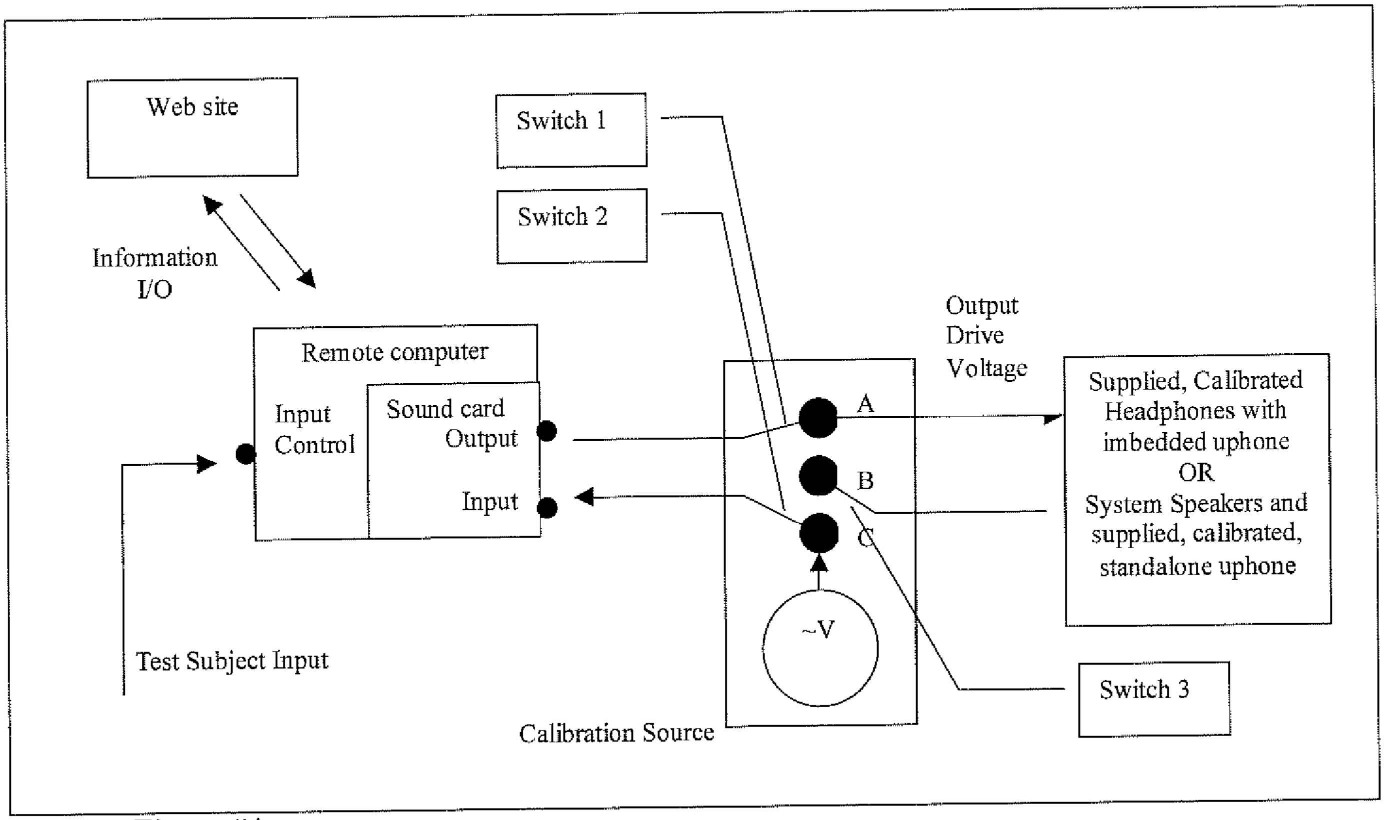


Figure #4

The VCO Method This method assumes that the sound card input port's frequency–to-frequency transfer function is one. Using this assumption, a VCO is used to map the calibrated microphone output voltage into frequency space. The microphone output is determined through frequency space analysis of the VCO output combined with the

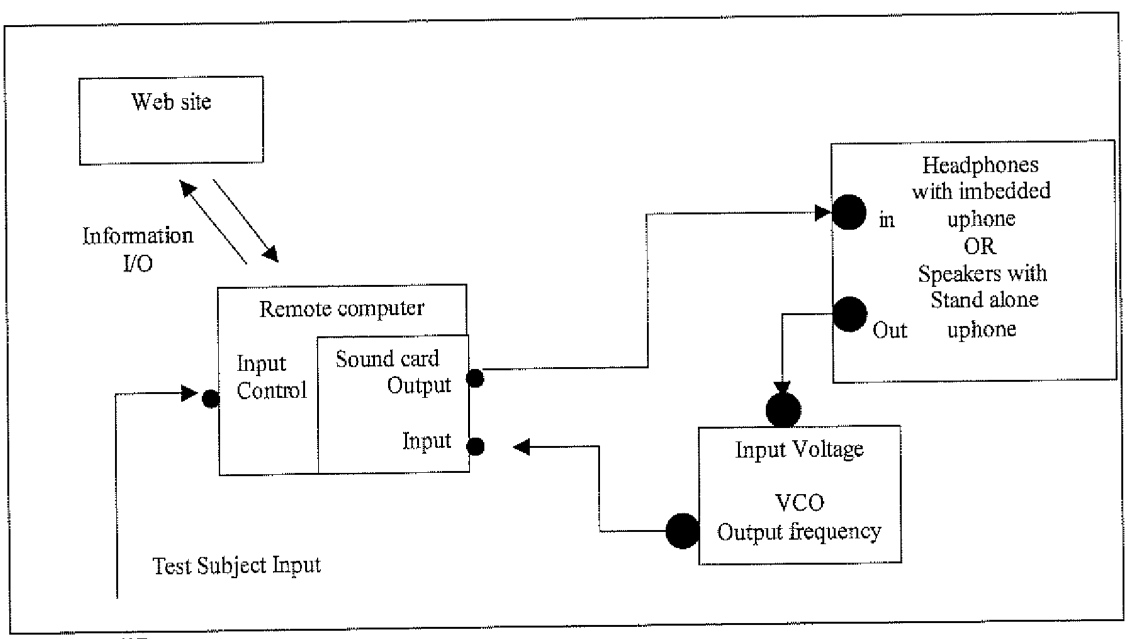


Figure #5

known microphone calibration. The test set-up is shown in figure #5.

In this test method, the test subject signals when the test criterion is met. Using the measured frequency of the VCO output, the known VCO mapping function and the known microphone calibration, the corresponding SPL can be determined. These data are then used to calculate hearing loss according the test type.

Relative Measurement

For some situations, the SPL corresponding to a test criterion may not be as important as knowledge of the relative threshold of hearing across the audible frequency range. The following section outlines relative SPL test methods. Each method described could be implemented with either headphones or speakers. These test methods assume that sound card's input A/D and its output D/A converters are linear across frequency and amplitude. Furthermore, the transfer functions of any headphones, microphones or speakers used in the test are also assumed to be substantially linear across frequency and amplitude.

In general, the test set-ups and protocols are simpler in a relative measurement. All the test methods discussed below can use the test set-up shown below.

Relative SPL Measurement Inferred from Drive Voltage

This is the simplest test method. No additional equipment is needed. The test proceeds by playing the test sounds either through the speakers or through headphones. The test subject signals when the test criterion is met. The measured data is the sound card's digital word to the output. The test set-up is shown in figure #6.

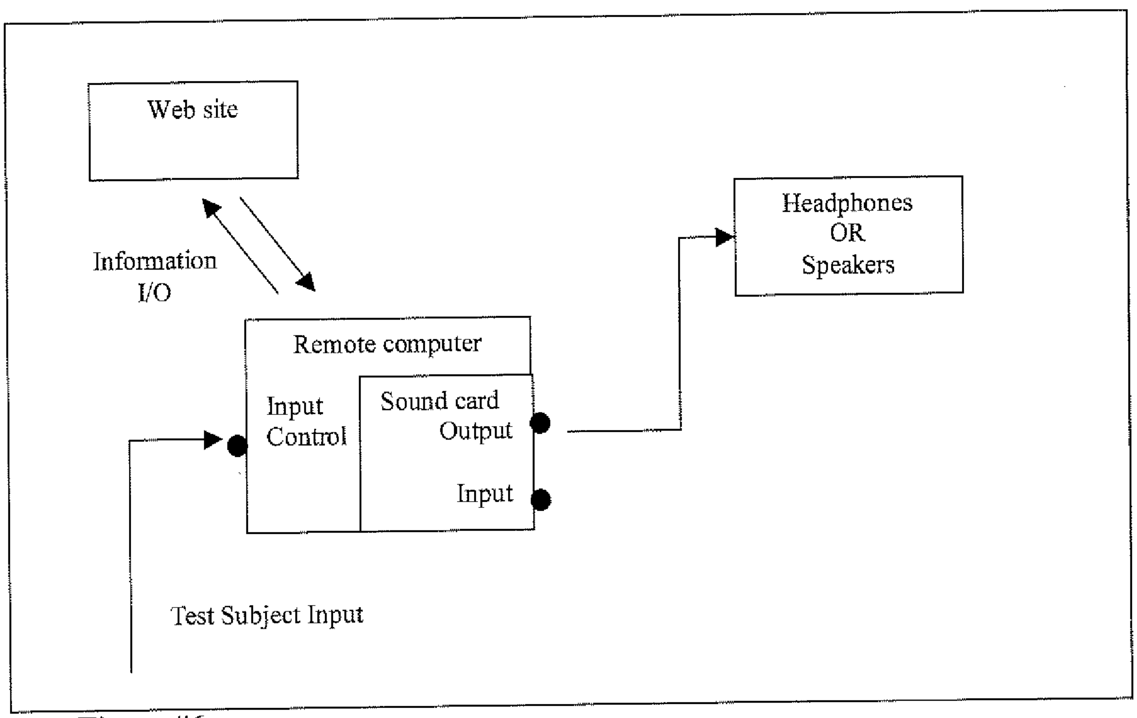


Figure #6

Actual SPL Measurement Completed at Ear

This method requires that a microphone be used to measure the actual sound pressure level at the ear. The test configuration is shown below. The measured data is the sound card's input digital work.

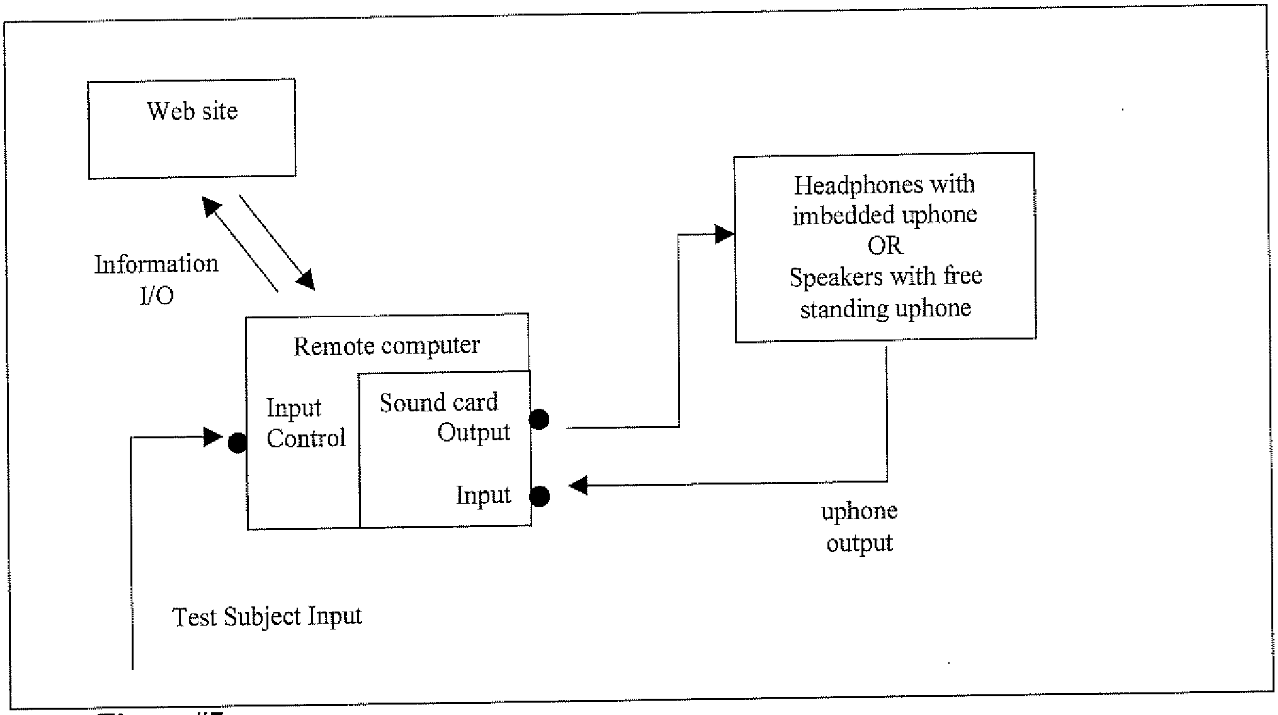


Figure #7

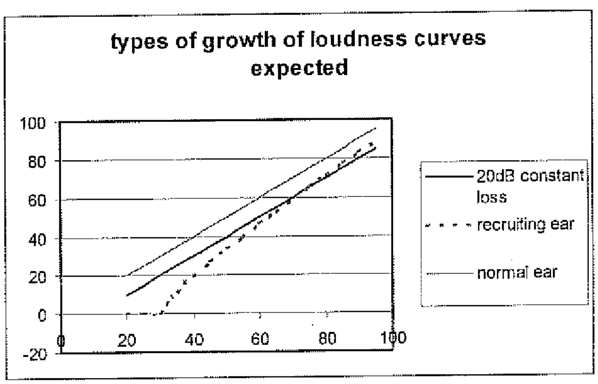
EXHIBIT B

Rx Sound
Personalized Listening

MEMO: Up-date on Progress of Loudness Matching Test Development
TO: Scott Huddleston, Sunil Puria Scott Rader, John Winstead
FROM: Chris Menzel
DATE: 2/4/00

The following document is meant to give a brief overview of progress relative to the development of a loudness matching test for the internet.

Attached are a number of curves that show growth of loudness. Each curve represents data collected for one frequency. A curves is generated by plotting the SPL value of the test tone that was deemed equal in loudness to the standard tone (1Khz) on the x-axis against the expected SPL level for that frequency at the corresponding 1Khz amplitude. The expected results of this type of plot are shown below:



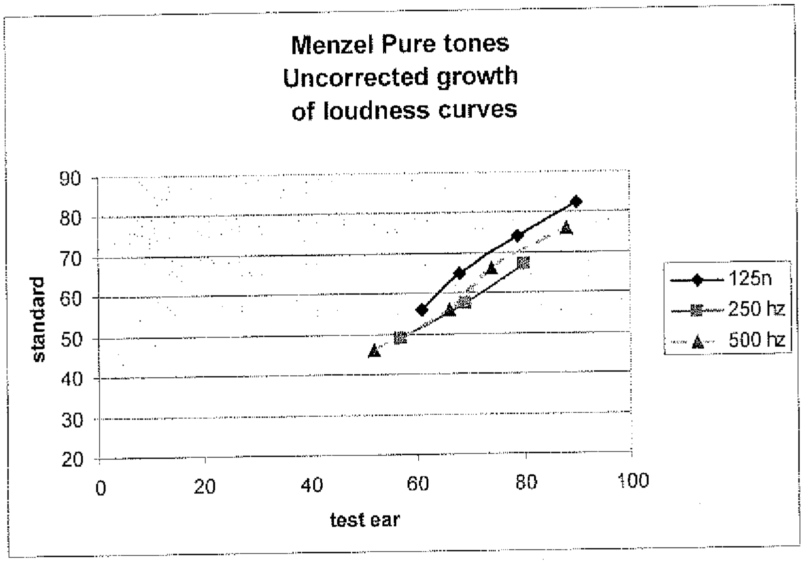
The hope is that using regression analysis the supra-threshold data can be used to assess threshold .

So far the following people have been tested with the following types of tones:
Chris Menzel: pure tones and noise
John Winstead: pure tones...noise to follow soon
Victor Samolis (my father in law...pretty severe hearing loss~log linear at 30dB per decade starting at about 500hz age ~65) noise
Clare Samolis (my mother in law...no known hearing problems age 69) noise
Christa Menzel (my mother ...no known hearing problems age 64)pure tone

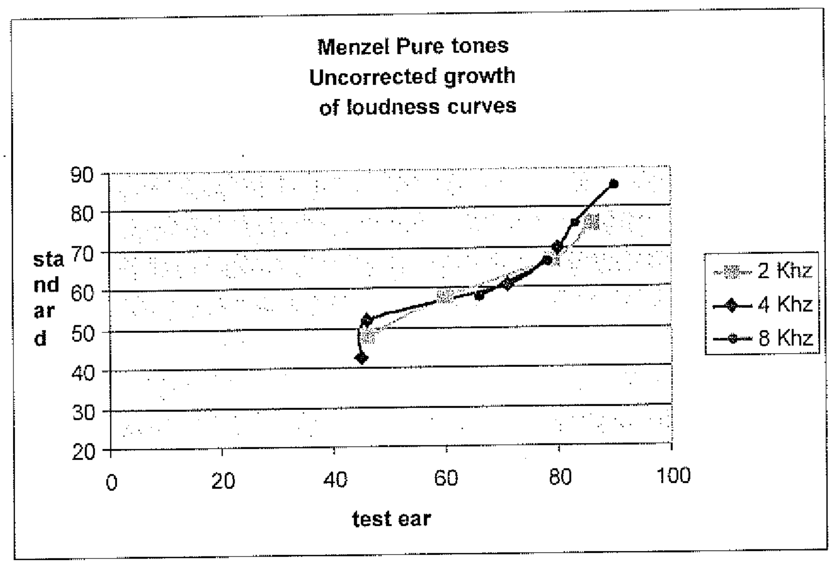
I will go through the growth of loudness curves for each of the above.

Chris Menzel (I have a slight best ear hearing loss 15dB at 6k and 20dB at 8k)

I've separated out the low frequency curves from the high frequency curves in order to see things more clearly and to separate those where I don't have hearing loss from those where I do. From the curves, one can see that the slopes are generally constant and that a linear fit would be pretty good. In fact, that is what one finds. Slopes are greater than .85 and less than 1.25. To my mind, the linear slope value and the "goodness of fit" shows that a linear model is appropriate and hence that I don't particularly have a hearing loss here. This result is consistent with my audiogram.



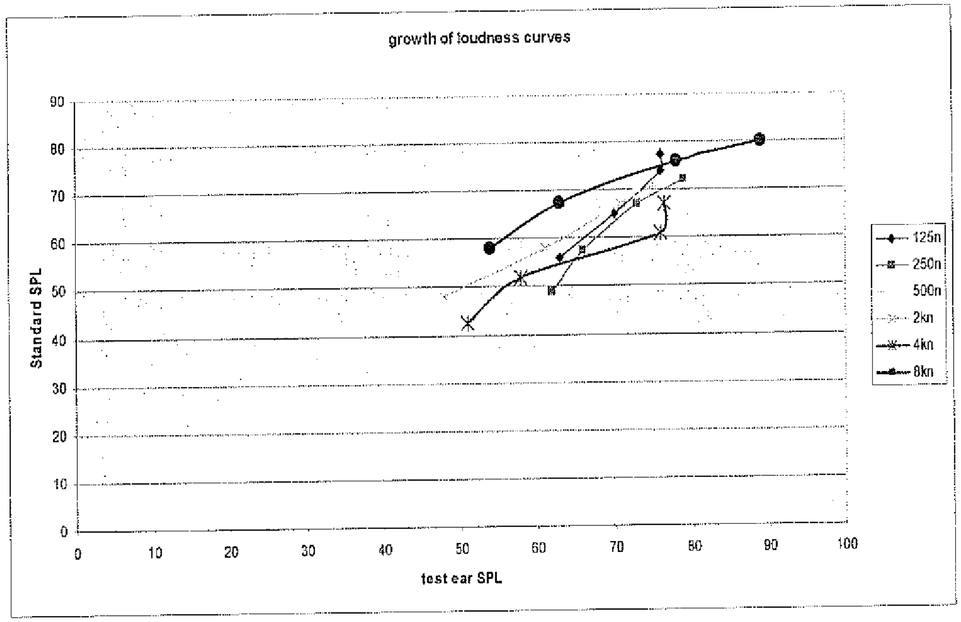
Now to look at the high frequency stuff. As one can see, these curves are not so clear cut. The general shape of the 2Khz and the 8Khz curves doesn't match expectations. Further there is a very funny wiggle in the 4Khz curve. It is difficult to tell what kind of model would be most reasonable here. I believe that part of the problem with these curves comes from the problems associated with using pure tones in a reverberant environment.



Now to move onto the noise pulses.....
Because of the way the data fell, I didn't need to separate these curves into high and low frequency traces for them to be visable. So there are all together. In this set of curve, the 125 hz, the 500 hz and the 2K hertz curves show a linear shape. This shape is consistent with my audiogram. The curve for 250hz looks abit like a recruiting ear however, it is flattening out to a slope much less than one at the higher SPL levels. Hence, this curve is a bit screwed up and its difficult to say exactly what model to use for it.

By contrast, the 8K curve looks tantalizingly good!. It intersects the 8Khz MAF (15dB) at about 30dB.

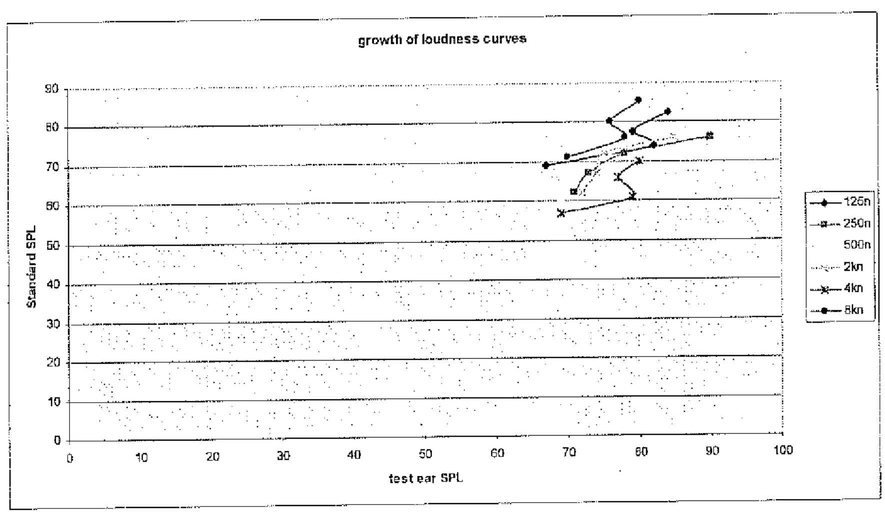
Note that the 30dB is a bit of an overestimate. In addition, the slope of the line at the higher SPL is again flatter than the expected 1. So its not so clear that this data is all that good either.



As can be seen from the curve, the 4Khz curve has a kind of funny shape to it altogether. Even if some of the data is left out, the predicted threshold will be way off(remember that at 4khz the MAF is –4dB)

In summary, much of this data doesn't look all that good.

Now to move on to John Winstead. His test was done using pure tones. He is doing a test with noise and the results will be analyzed shortly. To put it very simply, there seems to be a problem with the data. This is most clearly noticed by the fact that most all the curves (500 hz and 2khz excepted) have the same funny "bump" in them. I imagine that there was some sort of a problem with that whole set. I imagine that the problem is related to the pure-tones. Interestingly, we do have two sets of data for John Its tabulated below:

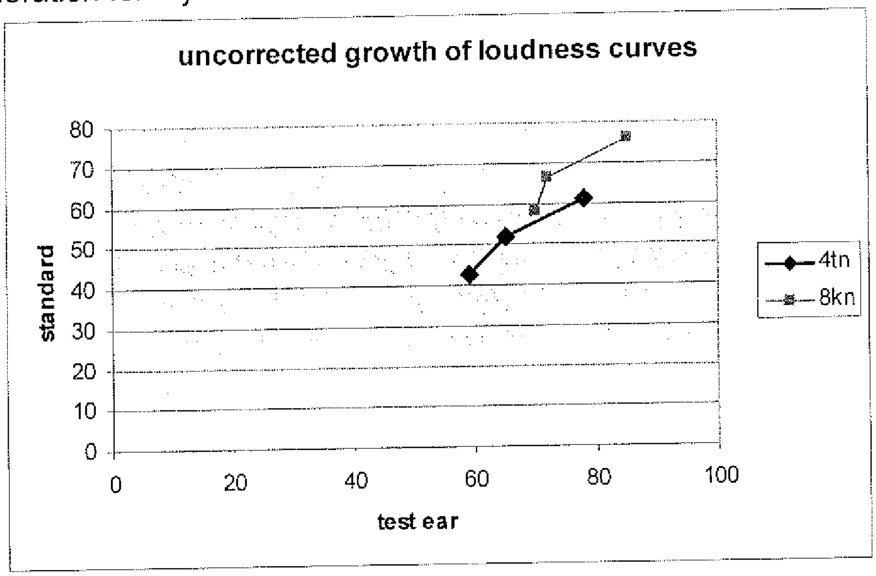


1Khz SPL	First test	Second test
65	72.5	70
70	85	78

75	90	76
80	95.5	89

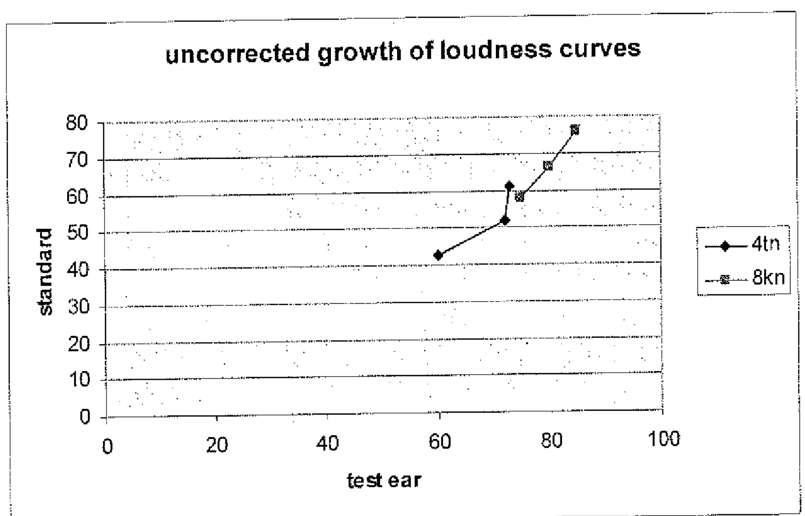
The data shows general poor repeatability.

Vic Samolis. Out of consideration for my father-in-law's time, I did not take his data at all locations. Instead I did 4 and 8 Khz and took one measurement at 500hz. Since the one 500 hz measurement was normal, I stopped.

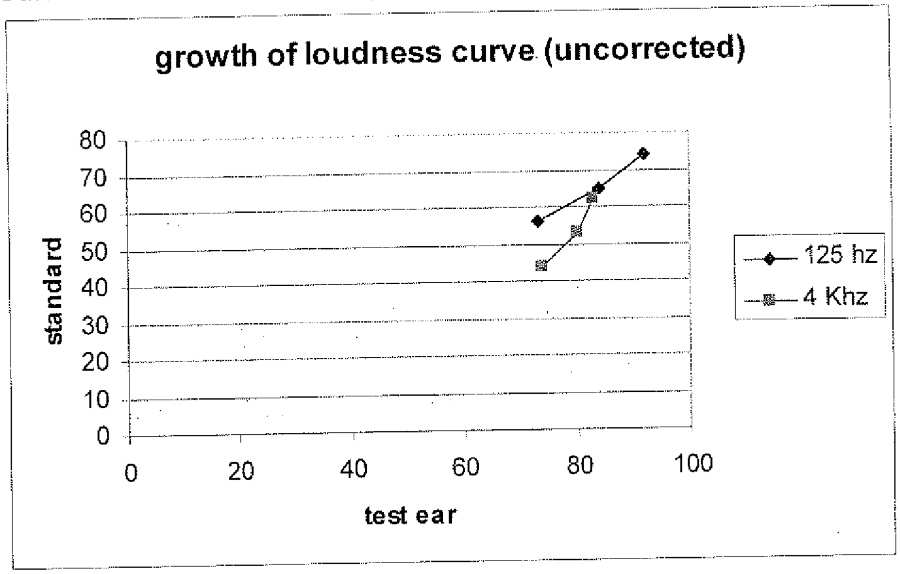


Now these curves show promise! One would imagine that the 4Khz curve would cross the the 4Khz MAF (-4dB) at about 45dB or so giving a threshold of about 50dB. This compares reasonably to the measured threashold of 60dB. For the 8khz data, the predicted threshold would be in the 60-70dB range. Again this values compares favorably with the 65dB loss measured.

Clare Samolis. Again in the interest of time, I only did the high frequencies. Here, since we do not know the extend of her hearing loss (though she does believe that she doesn't hear as well as she used to), what we can get out of the curves is related to whether they look like we expect them to and whether we believe in a general sense what is going on. The 8khz curve is generally reasonable in shape. Fitting a recruiting ear model to it would give a loss in the 60dB range. I don't know, but that seems awefully high for someone who doesn't have a big problem. Furthermore, the 4khz data has a funny shape altogether. Because the low SPL points tends to have a slope of one, one might argue that it's the higher SPL point that is out of wack.....but who's say for sure?(and of course that's the problem).



My mother's data (as I said, its not safe to visit my house). The 125hz curve looks pretty normal. In fact it would seem that there is very little hearing loss altogether. The 4Khz curve has the wrong inflection. That's a problems that seems too occur a bunch.



So that's what I see with the data.

The main problem that we are having is that a great many of the curves do not have a sufficiently normal shape to know how to model it (linear or power curve). What I'm concerned about is that the quality of the data is insufficient to allow us to distinguish stuff.

<u>EXHIBIT C</u>

Document Revision History			
REV	Change Description	Originator	Date

RxSound
Personalized Listening

RxSound Corp,
801 Welch Road
Palo Alto, CA 94304

	Name	Date
Original Author		
APPROVALS		
Approver		
Approver 2		

Document Title

AudioTest
Software Requirement Specification

Document Number

Xxxxxxxx70

Revision

REV

1. Introduction

1.1 Purpose

AudioTest is a small application created to prove the concept of performing an audiometric test using a standard PC and calibrated head phones.

In its first implementation it will facilitate easy manual administration of tones to subjects along with recording of the results in some sort of datafile.

Please see chapter 2.4 with respect to future developments.

1.2 References and Glossaries

[ITHFS]	Internet Hearing Test Functional Specification. Chris Menzel.

1.3 Document Scope

At this point the document is a work in progress to ensure that the developer (Benny) and the project manager (Chris) have a mutual understanding of what the program should be doing.

2. System Overview

This section describes the program and the environment in which it must work. It also presents a short list of the main features to be incorporated.

2.1 *System Description*

The AudioTest program is a standalone PC program. It communicates with the user through the screen, mouse and keyboard. It communicates with the sound card to produce various stimulus as output, and maybe obtain some sound level measurement as input.

2.2 *Key Functions*

The program has a number of user related functions:

- Calibration. The user can follow a step by step procedure to calibrate the system.
- Enter client information. A very simple set of fields allows the user to identify the patient under test.
- Administer test. This function allows the user to use the screen as a simple (yet flexible) audiometer.

2.3 *Program Limitations*

This development will NOT include an installation program, neither will help files be included. It is only an "in house" program.

For the first version, the user interface will be functional, but not really "fancy".

It is expected that the program can use standard Windows functions for controlling the sound card (if not it would be a nightmare keeping updated with different hardware configurations).

Also see section 4.4 – Software Interface.

2.4 *Program Future*

The program is only a "beginner program", intended to achieve two goals.

- Provide Chris a platform on which he can refine the hearing test and provide the audiologists a better tool to administer the test.
- Give Benny a first introduction to the problem domain.

Through gradual development, the following features are expected to be added:

- Automatic test administration. This includes implementation of one or several test algorithms.
- User interface improvement.
- Gradual componentization, so the core parts of the test can be used in several development environments. (Visual Basic, Java, JavaScript)

- Eventual transition so it can run on the web – on a yet to be determined set of platforms.

2.5 *The User*

The first version of the program will only be used internally in RxSound.

2.6 *Prerequisites*

None.

3. Functional Requirements

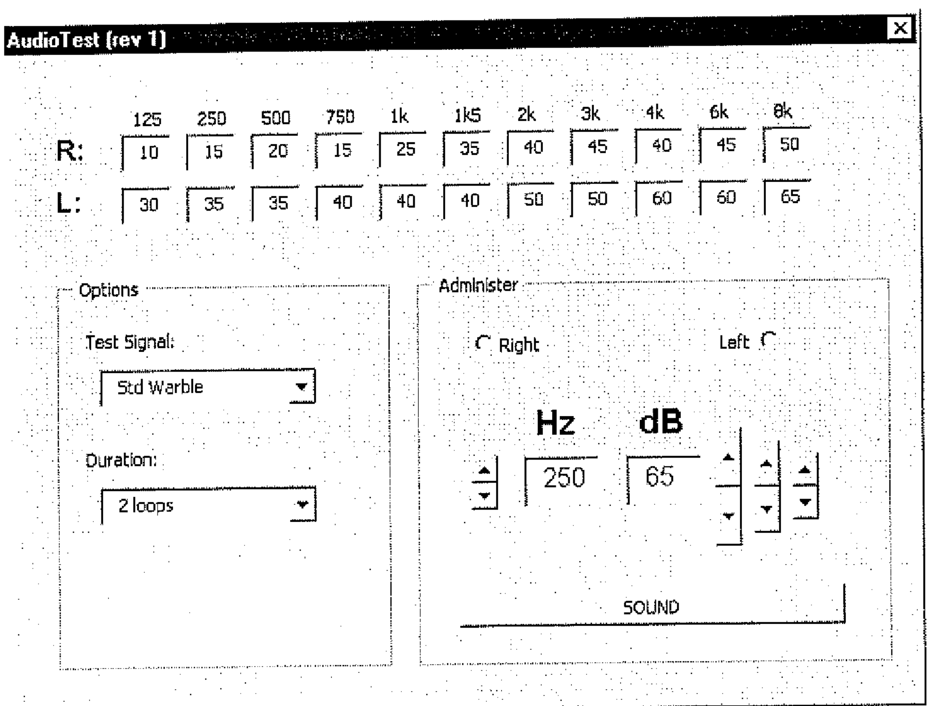
This lists the requirements from the users perspective.

3.1 *User Screens and Menu(s)*

There will only be one screen, looking as the one shown below. By accessing a menu, the user has the following options:

- New: Creates new file, ready for
- Open an existing file (which allow the selection of a datafile)
- Save. Save the measurement
- Save as.... Save the measurement under another file name.
- Calibrate.... Opens the calibration dialog.

3.2 *The AudioTest main Screen*



The AudioTest screen is divided into 3 sections:

3.2.1 The Audiogram Display:

This area shows the recorded values. It is read only, and will be updated as the test progresses.

3.2.2 The "Administer" Area:

These controls are used to administer the test. Using the up and down arrows the user can change the frequency or the sound pressure level. Pressing the largest button changes the sound pressure level in increments of 5. Pressing the smaller ones changes the level in increments of 3 and 1 dB respectively.

To initiate a signal the user will press the SOUND button, and a number of things may happen, depending of the settings of the "duration" field in the "options" section.

- If duration options is : "no loop" the sound will stop as soon as the button is released
- If duration options is: " x loops" the sound will sound for x loops of the sound file.
- If duration options is "indefinitely", the sound will continue until the button is pushed again.

3.2.3 The options area:

This should allow the "expert user" to try out a few things, that are then reflected, when the actual administration is performed.

Currently two options are suggested:

- Test signal. The user may select different types of test signals. What this boils down to in the end, is that different families of wave files are being selected. *QUESTION: Do we really want this here? Is it not a mess if part of the test is conducted using one signal, and other parts are conducted using another? So maybe it should not be possible to mix signals this easy.*
- Duration. This determines how long the signal is administered (also see above).

4. Interface Requirements

The following is a list of requirements to the user interface, as well as to interfaces required for this software to work.

4.1 User Interface

N/A

4.2 Hardware

At this point a PC with Windows 95 or above, and sufficient Sound Card capabilities as determined by Chris.

4.3 Communications Interface

N/A.

4.4 Software Interface

It is anticipated that the whole program can be written only using calls to standard Windows API, or maybe the DirectX library.

Chris will deliver to this application a number of different sound-files, in a yet to be determined format. AudioTest will be able to read these files and use them to generate the test stimulus after some minor modifications. Modifications include only:

- Attenuation
- Looping (a set amount of time)

(The whole sound generation will however be abstracted, so if things have to be done differently at a later stage, this should not affect the user interface or the test algorithm significantly)

5. Other Requirements

I am sure I can think about something.

5.1 Timing (Real time Requirements)

None – other than what is required to support the production of sound.

5.2 Quality

Please rank the following attributes in order of importance:

- At this point – development time is most critical.

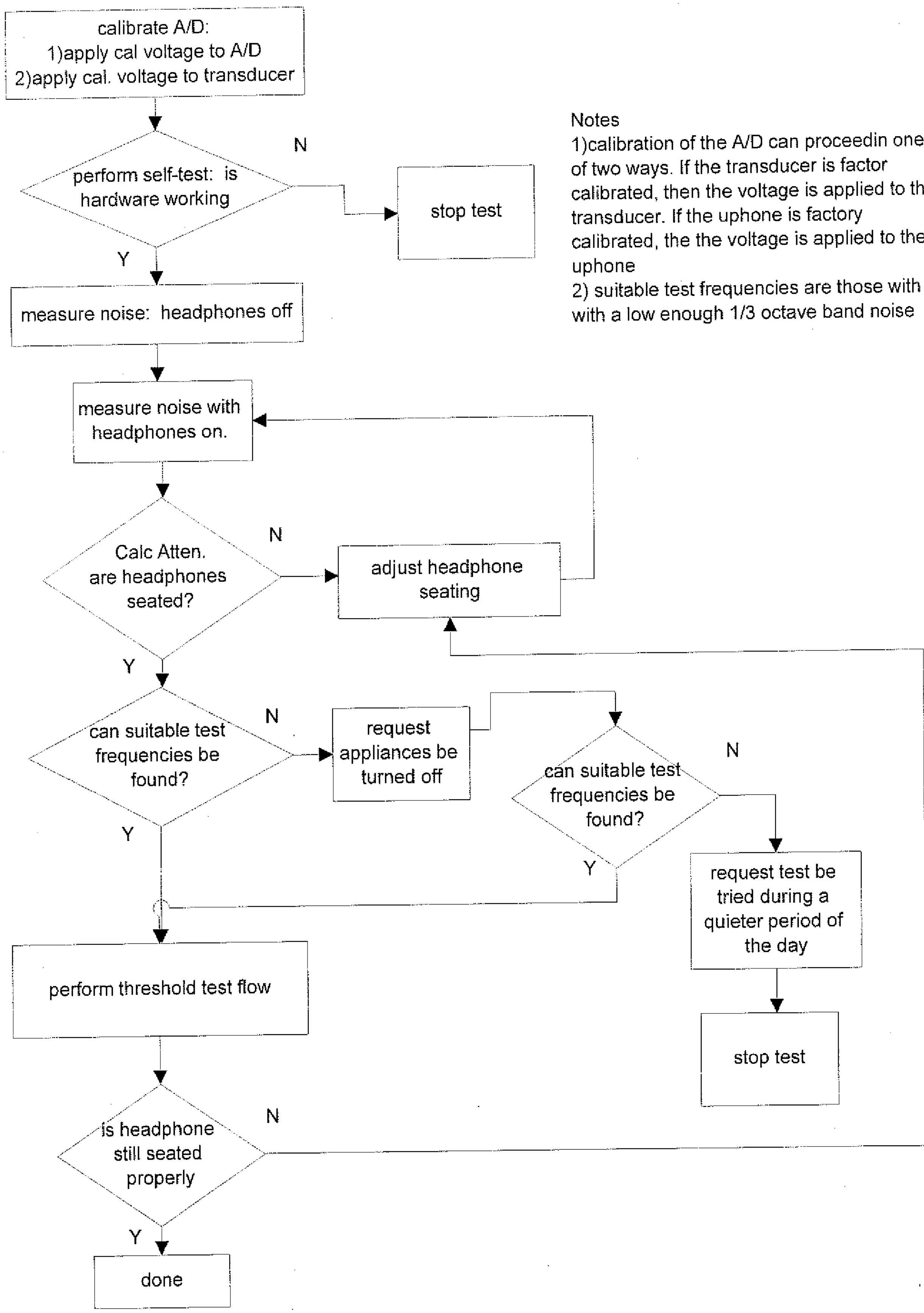
5.3 Data logging

The result of every test will be logged in a .tst file. The format and contents of this file is still TBD.

5.4 Outstandning Issues

N/A

EXHIBIT D



Notes
1)calibration of the A/D can proceedin one of two ways. If the transducer is factor calibrated, then the voltage is applied to the transducer. If the uphone is factory calibrated, the the voltage is applied to the uphone
2) suitable test frequencies are those with with a low enough 1/3 octave band noise

Some things we've discussed wrt to implementation:

Calibration of A/D. There are (at least) two methods to do this:

Method	What knows	Where is cal signal applied	comments
Uphone based	Uphone sensitivity and frequency response	Signal is a voltage, it is applied to the Sound card A/D	
Transducer based	transducer sensitivity and frequency response	Signal is a voltage, it is applied to the tranducer	Full sound field generated must be known ie are headphones on head?,what reflections in room etc

Determination of which one is best will be related to:

- Robustness of calibrated component: how likely to change during shipping and on site
- Cost of calibration per component OR cost to perform calibration

Headphone leakage

This issue may be less of an issue than previously represented. We have a paper that shows that for holes leaks up to 1.65mm diameters, the effect of the leak has disappeared by about 400hz.(Anderson and Whittle, Acustica 1971). So it can be surmised that larger leaks will have little or no effect at 2K and above where we expect to make our measurements. Clearly some testing will need to be done on this issue.

Background noise level measurement

All of our estimates on what the background noise inside the headphones will be (and hence our smallest measureable hearing loss) are based on the noise present in a 1/3 octave band. Indeed, it is assumed that the total room noise will be more in the 50-60dB range. Hence, it is a necessary condition for us to actually measure the 1/3 octave band noise. This does two things:

1)changes how we look at the uphone noise specification ie we will need to know the 1/3 octave band noise as a function of center frequnency

2)requires that spectral analysis capabilities are part of the system.

EXHIBIT E

RxSound
Personalized Listening

Memo

To: Scott Rader; Chris Menzel; Sunil Puria: John Winstead

From: Benny B. Johansen

CC:

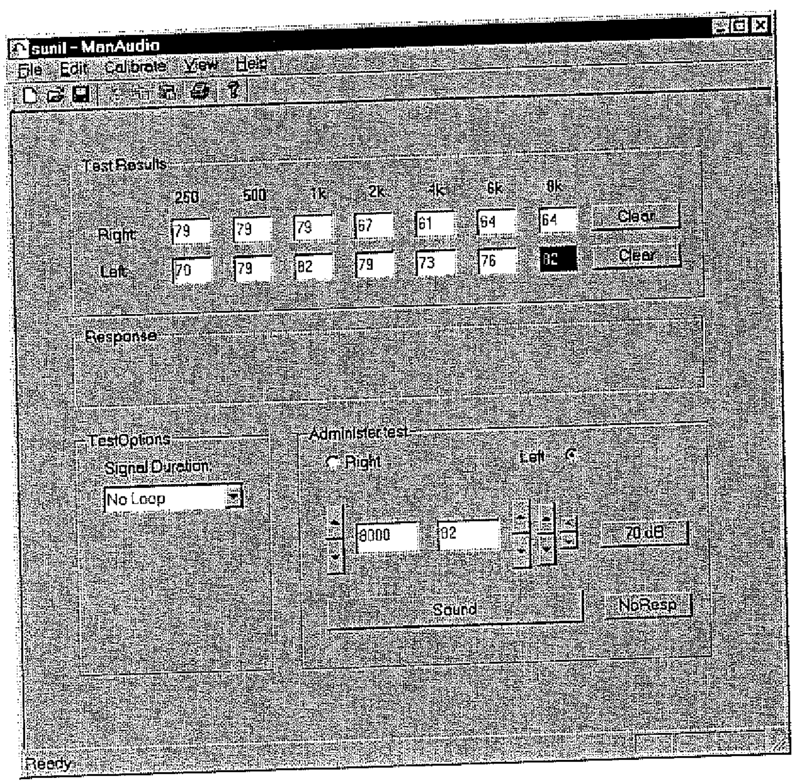
Date: [redacted]

Re: Status and future plans: Update on Manual Audiometer Program

Purpose and Contents

I have now created the first visible program, which tests out some of the features which we need in the home audiotest and for future research. Below I will first mention where I am with the audio test, then I will outline my plans on the development front for the next couple of weeks.

Status: The ManAudio.Exe program



How to Find and Install?
Currently it is available for installation on RPA2\RXTRANSFER\MANAUDIO. Just run the setup program and follow the instructions.

What happens the first time/Calibration?
When you run it the first time, you will probably get an error. This is because the program relies on calibration data, which is specific to your sound card. To "calibrate" your system do the following:

1. Notice the key, which the program claims it is missing. It will have the format SC_XX_XX_XXXX
2. Connect the calibrated phones and locate the SPL meter over one of the loudspeakers.
3. Use the function Calibrate|1000Hz at peak output, and note the output on the SPL meter.
4. Add an entry: SC_XX_XX_XXXX= <measured value> to the file ....manaudio.ini under the [SOUNDCARD] section: as shown below:

```
[SOUNDCARD_SPEC],
;SC_<ManufNo>_<ProductID>_<DriverNo>          =          <MaxOutput>
;ESS                                                     Technologies....
SC_46_1_200=93
SC_46_2_200=0
SC_1_100_500=93
```

Close the program, and restart it, and your system should now be "calibrated".

Notice that if you later install ManAudio again this file will be overwritten.(Not nice, but it will take too much time to fix for now)

Functions to measure threshold:
The function is modeled an audiometer, and you should be able to use it without using the keyboard.

<key up>/<key down>:	Level up or Level down in 3 db steps. Hold down the left <shift> key and you will increase and decrease by 6 dB. Hold down the right <shift key> and you will increase and decrease by 1 dB.
"7"	Set level to 70 dB
"N"	Mark "NoResponce"
<SPACE>	Present Sound

The values shown on the screen are db HL.

If you locate the mouse on top of the field named "response", <right mouse clicking> will cause the "response field" to light up, indicating that the user could hear the stimulus. (An entry will also be written in the log file).

File Management:
Very simple. Use "New" to create a new test, and Save and Save as to save test results.

The result will be saved as an .aud file, and notice that this is a text logfile, that should be viewed using notepad or any other text editing program.

The LogFile:
The logfile is separated into the following sections:

#General	Standard information about the test.
#DATA	The actual measurements
#HARDWARE	Identification of the wave in and wave out hardware and software drivers
#LOG	A fairly elaborate log of the flow of events, including the users response.

What has been tested

At this point I have only installed the program on my own and Chris Menzels machine. It would be interesting to try it out on more machines, to get a good feel for the capabilities of the different sound cards and my ability to control them.

Which capabilities have been demonstrated?

- The overall architecture of different modules communicating via formatted text strings seems to work well. Likewise the logging seems to be useful.
- I am able to control the sound card as follows:
  - Play any wave file, implement looping and attenuation. Relate output to a calibration factor.
  - Capture wave input, simultaneously with playing wave output.
  - Increase the controls on the audio mixer when running the application, and resetting them once the application is completed. (Currently I am only doing it for waveout and overall volume out. I also know how to do it for microphone in, but it is a rather lengthy procedure, so I have left it out for now).

Which capabilities are (almost) also available?

- I have a separate program, which does FFT on the captured input signal. I just have not figured out how to calibrate the input. (Once I have, I think it would make sense to include another row of numbers on the audiometer showing the measured noise floor at each of the target frequencies).
- Same program also demonstrates a little graph utility which plots the results.
- Whether you select "left" or "right" the sound will be presented binaurally. I know how to change this to monaural presentation, I just have not had the time to do it.

An Interesting Observation:

Although the whole installation "disk" is currently 1.5 MB, it is interesting to notice that the program itself is only 76 KB. The rest is taken up by the wave files. Once we settle on the type of wavefile we want, it would be fairly easy to write to the code to generate those files internally, leaving the total program to be less than 100 KB, something which is certainly possible to download over the internet.

The next couple of Weeks:

I think the ManAudio application can be useful for testing a number of things with respect to sound cards, microphones etc. and I will continue to extend it, whenever it make sense.

However the ultimate demo, which I am working towards is the following:

- User accesses a web-page on our internal server a) using IE4 or IE5 b) maybe using Netscape 4.0 or higher.
- The page contains a hidden activex control and a single button, which can be used to indicate if a sound can be heard.
- Using this page, the user can take an interactive hearing test, wearing our calibrated head phones.
- Once the test is completed the results will be stored on our central database server.
- The user will then go to a second page showing a list of music selections.
- The user can pick one selection. This selection will immediately be processed in the back end by a PC with a DSP board.
- A new screen will then be shown to the user containing a link to the modified sound file.

I plan to take the following approach:

- Write a simple C++ program which implements the automated procedure. Have John Winstead try it on a few people.
- Change this simple C++ program into an Active X control and embed it into a web-page
- Have Scott find some DSP consultants to implement the compressing routine on a Texas Instruments DSP processor. If this cannot be done in time, I may just implement something very simple to demonstrate the principle, or "borrow" Brian Moores algorithm and run it on the server.
- Implement the link between the work station Active X control and the back end server.
- All work will be done on Internet Information Server 4, and using ASP + one of two COM objects.

With a lot of luck and some help I can probably accomplish this by the end of April, beginning of May.

EXHIBIT F

R Sound
Personalized Listening

Memo

To: Brent Edwards, Chris Menzel, Scott Rader

From: Benny B. Johansen and Sunil Puria

CC:

Date: 05/26/00

Re: Patent Disclosure

Purpose and Contents

The following document briefly captures thoughts and ideas developed during the month of April, 2000 regarding our ability to control a sound generator, a sound filter or any other sound processing device from any computer by the use of a light source, electro magnetic emission originating from the computer, through a PDA port (i.e., a PalmPilot) or through a mobile phone port.

Why is this beneficial?

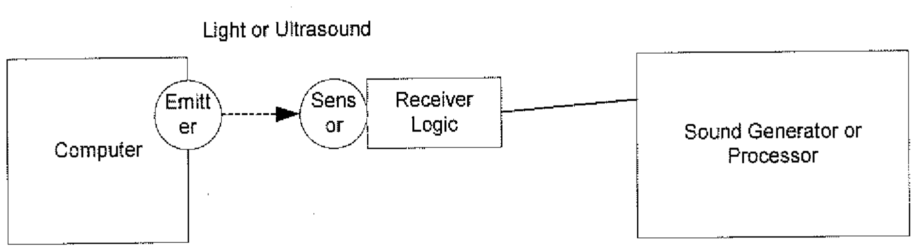
Some of the advantages are:

- Using the described method you can control any of the above mentioned devices from a computer, without having to rely on an electrical connection between the two devices. Such a connection may not be available (printer port, game port USB port etc.) on all computers. In addition, a connection which may not always be available (like sound card, printer port, game port etc.). Further such a connection is potentially hazardous and could cause electrical shock.

- Light and ultrasound are both inaudible and will therefore not interfere with any hearing test or sound processing going on simultaneously.

- Connecting to a mobile devices that allow measurement of hearing loss through portable devices rather than being connected to a computer. The hearing profile can then be used for a number of different appliances that might be connected to these devices.

What are the methods suggested?



Using Light
The sensor would be one of the following:

- Normal photo transistor
- Small CCD cell

The emitter would be:

- A small portion of the computer screen, be it an LCD, TFT or TV screen. A number of methods could be employed:
    - Simple switching on and off of pixels to generate some kind of Pulse Coding
    - More complicated patterns like the flashing of bar codes, or some of the new security (stamp type codes) at regular intervals
    - Employing the scheme currently used by Timex to download information to a watch.

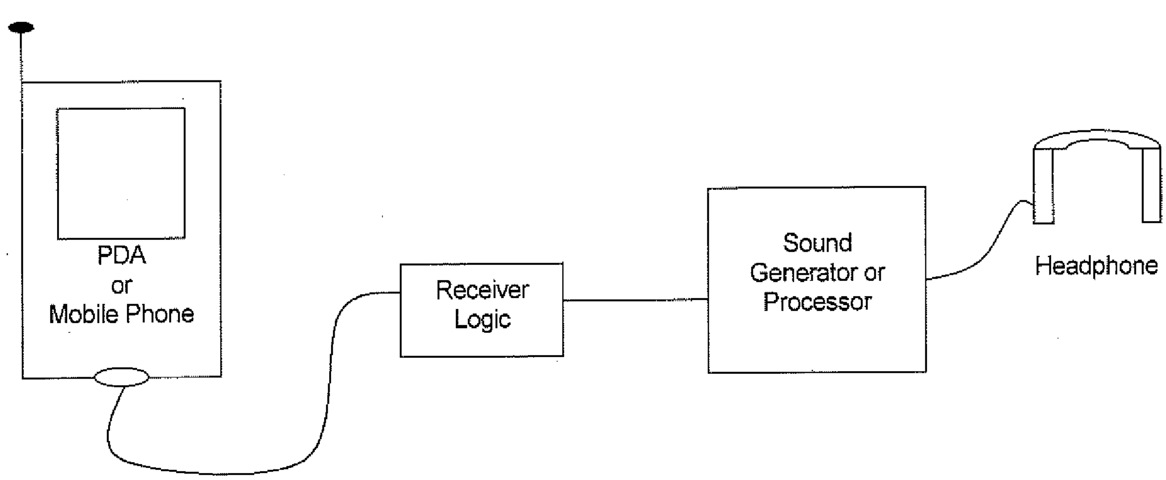
  At this point we envision that a user would be given a small receiver – the size of a rubber and asked to place it a designated position on the computer screen. (guided by a logo generated by the software running). Once the receiver was correctly placed the computer could manipulate the pixels in this area, and control the connected processor.

- Another light source is obviously the IR port already available on many computers, PDA's etc.
- A third light source would be the hard-disk light. I bet that on 90% or all computes it is possible to get access to this light and switch in on and off very rapidly, just producing control information.

Using Ultra-Sound:
The sensor would be a standard ultrasonic receiver or maybe a magnetic coil.

The emitter would be the loudspeaker. We are guessing that even if the loudspeaker will not be able to transmit the sound, a sensor coil would maybe be able to pick up the electro magnetic emission from the loudspeaker coil, if the two of them are placed close enough.

Mobile environments
Hook up the above described headphone to a Palm Pilot, or any other PDA, through the port at the bottom. PDA are increasingly being used are wireless devices that are hooked up to the internet. If our web site is designed to communicate to these PDAs, then we can send bits of information to the



headphone to do a hearing test connected to a PDA and get user information back from the PDA.

Similarly, mobile phones are increasingly being used as mini-web browsers. Again, we should be able to hookup our headset to the cell phone port so that it communicates with the decoding circuitry in the headphone and thus allows us do a hearing test with a mobile phone. Note that in this case we are not using the mobile phones receiver. We are using our headphone to do the test.

The hearing profile assessed with the mobile devices can then be used by various appliances that might be connected to the mobile system. An example of such an appliance is a speaker phone for the mobile phone installed in an automobile. Another example is if the PDA is equipped with the ability to play MP3 files over the internet then the MP3 player equalizer settings could be adjusted to match the hearing profile of the individual.

All of the above measurement methods are intended to allow us to make hearing threshold measurements. With more complicated timing circuitry it is possible to design a loudness balance test into the system. The equipment is not intended to have the capability of playing music.

EXHIBIT G

From Page No. i-T hrs 7/4/00

6/4/00 patent search

automotive & radio all years 2935

AND volume 809

radio gives too many radio wave for com

So I'll try automotive & audio → 1530

& Volume → 462

+ & "volume control" → 110

11-4/1999-2000 only

the apparently important patents

[redacted]

7/6/00

Sunil commented that he heard a distinct click @ the shutoff of the

To Page No.

Witnessed & Understood by me. Date | Invented by | Date

Recorded by

masker in the 4kHz forward masking
so, I'm looking into it.

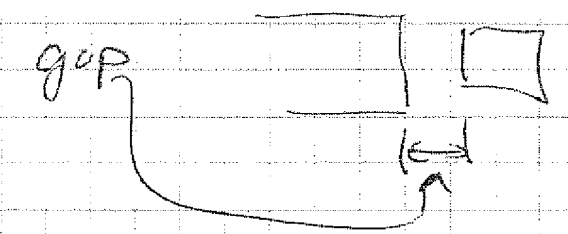
I find I hear the clicks on my machine
here in CT but don't recall hearing them
~~in~~ CA, seems like a sound card problem!
I lengthened the turn off time (raised
cosine of 50 HZ, no click.
so lets try some other shutoff times
100 HZ nothing distinct
200 HZ may be starting
400 HZ definitely hear it.

Sunil tried listening on my machine
at CFI and got very different results than
he got on his machine —
he didn't hear the pulse turn off clicks
~~out~~ until he used headphones. He used
the line out (not speaker out) with the
headphones

Witnessed & Understood by me, | Date | Invented by | Date
Recorded by

n Page No.__

8/2/00
forward masking using TDT equipment
Signals are from 4k-d20-p20a.sig
test is 4khzpa. PA



10, 20, 30, 40, 50, 60, 70 usec

Done with headphones (no HB7 Headphone drive atten)
on right ear

SGT	gap	mean	std	comments
1	10	-38.3	1.6	
2	20	-66	3.58	
3	30	-77	2.5	Probably represent noise floor since lower limit of signal was -110
4	40	-78	5.3	
5	50	-77	2.73	
6	60	-78	4.38	
7	70	-76	2.12	

Calibration was such that there dB's are relative to ~90dBSPL i.e -77dB => 13dBSPL

noise floor ~ 60dB
(weighted-averaged

tried doing this same thing w/ Peltors —
there seemed to be some sort of
sound artifact with the peltors —

To Page No.____

nessed & Understood by me, | Date | Invented by | Date
Recorded by

From Page No.

it was as if there was a very faint sound of glass breaking/tickling in the interval with the probe tone. it was ~~exp~~ great for getting interval correct. Those headphones seem less sensitive than the calibrating ~~[illegible]~~ the Sennheiser calibration up

$$\boxed{\text{Sennheisers} \quad 1\,mW \rightarrow 99\,dB}$$

doing calibration

$$\text{Power} = \frac{V^2}{R}$$

$$\frac{SPL}{Vrms} = \frac{10^{\left(\frac{dBSPL}{20}\right)}}{Vrms}$$

$$\frac{SPL}{mW} = \frac{SPL}{Vrms} \cdot \frac{1}{SQRT(\text{Power in mW})}$$

So for the Sennheisers

$1\,mW \rightarrow 99\,dB \rightarrow 89125\ SPL$

$$\frac{SPL}{Vrms} = \frac{SPL}{mW} \cdot SQRT(\text{Power in mW}) = \frac{89125\ SPL}{Vrms}$$

Witnessed & Understood by me, | Date | Invented by | Date

Recorded by

if the impedance is 150 Ωs,

Then $1mW = \frac{V_{rms}^2}{150}$ $V_{rms} = .387$

$V_p = .548$

This gave ~~@~~ ·101dB

So change cal to ~~@~~ 94dBSPL → .43V

for the Peltors

85dBSPL @ .43 Volts
~~@~~250 Ωs

$$85dBSPL \rightarrow \frac{\left(.43 \times \frac{1}{1.4}\right)^2}{250} = .377mW$$

.377 mW × 2.65 = 1mW
→ 8.47dB

So 85+8.47 93.5dBSPL @ 1mW

Now, with system calibrated
re-run test

	10	20	30	40	50	60	70
ave	41	31	32	31	.31	31	33.13
stdev	5	4	4	5	4	5	4

my absolute threshold for the probe is ~30-32

Project No.______

Book No.______ TITLE ______

From Page No.__

if 1 Volt = 94 dB $= 20\log(2^{16} \times .5) - 20$

$90 - 20 = 70$

~~1 volt~~ 94 dB is 70 dB bits

thould 31 dB . ~~66~~ 20 dB bits

.66 16 bit d

So here's the problem!

~~FA~~

Want to do the following ~~spec~~ products

* headphone for cell phone
to do cal. test WAP/regular cell

design review next week

So... This week make proposals

To Page No.__

Witnessed & Understood by me,	Date	Invented by	Date
		Recorded by	